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Converging Voice, Video and Data in WLAN with QoS support

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***Abstract*— Providing Quality of Service (QoS) for converging traffic in wireless local area network (WLAN) is still a major issue for the researchers and industries. Due to the limited bandwidth and exposure to unpredictable interferences in a wireless environment, efficient management and optimization of the access protocols are essential to provide acceptable QoS for real-time traffic and fairness to best-effort traffic. In this paper, we investigate the throughput and delay performance of the legacy IEEE 802.11 and the 802.11e when carrying converging data in WLAN. We evaluate our recently proposed protocol known as wireless token network protocol (WTN) and show that WTN successfully increases performance of converging traffic in WLAN by, first, decreasing the size of the frames and subframes of the MAC header, and second, by implementing centralized polling to reduce collisions of packets.**

I. Introduction

Wireless local area network (WLAN) is growing rapidly both in the private and public sectors. There is also increasing awareness by enterprises that their wireless infrastructure needs to allow multimedia applications to be added to the existing network data services [1]. Efficient management of access control for different types of traffic is essential to provide quality of service (QoS) to real-time traffic in multimedia applications while minimizing suppression to best-effort traffic (e.g. FTP).

The legacy IEEE 802.11 standard [2] has been widely used, but lacks QoS support for real-time traffic. The new WLAN standard known as the IEEE 802.11e [3] enhances the QoS of the legacy 802.11 and introduces priorities for traffic types to overcome some QoS issues for real-time traffic. Although the 802.11e supports some degree of QoS, many of its optimization features are left to vendors to design and implement.

In this paper, we investigate the performance of converging voice, video and data traffic in a wireless LAN using the legacy 802.11 and 802.11e MAC protocols. We evaluate a new proposed centralized polling MAC protocol, WTN, to improve the performance of this traffic convergence. We compare the results of our protocol with the existing protocol and show that our protocol performs better than the legacy 802.11 and 802.11e protocol.

A. The Legacy IEEE 802.11

The legacy 802.11 standard operates in two modes, contention free period (CFP) mode and contention period (CP) mode, known as point coordination function (PCF) and distributed coordination function (DCF) respectively. PCF is based on centralized polling while DCF uses a carrier sense multiple access/collision avoidance (CSMA/CA) medium access control (MAC) protocol.

In DCF, each node senses if the medium is idle for a period called DCF inter-frame space (DIFS) before transmitting. If the medium is idle for at least a DIFS, the wireless node is allowed to transmit. If the medium is busy, the node then enters a back-off procedure where a slotted back-off time is generated randomly from a contention window (CW) size.

Initially the CW is set to a minimum value, CW_{min} and doubled after each unsuccessful transmission attempt until it reaches a maximum value CW_{max}. If transmission is successful it is reset to CW_{min}. The back-off time is decremented by one slot when the medium is sensed idle for a DIFS. It is frozen if the medium becomes busy, and resumes after the medium has been sensed idle again for another period of DIFS [2]. Collision of packets occurs if the *CW* back-off time of two or more nodes reaches zero at the same time. A positive acknowledgment is used to notify the sender that the frame has been successfully received. If an acknowledgment is not received within a time period of ACKTimeout, the sender assumes that there was a collision and schedules a retransmission by entering the back-off process again until the maximum retransmission limit is reached. Legacy 802.11 also provides a mechanism to handle hidden node problems with a four-way hand-shake scheme known as request-to-send and clear-to-send (RTS/CTS).

PCF uses an access point (AP) as point coordinator to manage polling to the wireless nodes. With PCF enabled, the channel access time is divided into periodic intervals called beacon intervals which is composed of a CP and CFP. In PCF, an AP maintains a list of registered nodes and polls them according to the list. Nodes can only transmit when being polled and the size of each data packet is bounded by the maximum MAC packet size of 2304 bytes. PCF uses a

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shorter inter-frame space than DCF inter-frame space (DIFS) known as PCF inter-frame space (PIFS) as shown in Figure 1. One major problem faced by PCF is the link adaptation ability of the physical layer which supports multirate and makes the transmission time of a packet variable. Since the legacy 802.11 has been comprehensively explained in the literature [4] [5] , we omit further details of it.

B. The IEEE 802.11e

The 802.11e standard defines a superset of features specified in the 1999 edition of the legacy IEEE 802.11 MAC protocol [3]. It introduces two main functional blocks, the channel access period (CAP) and traffic specification (TSPEC) management. CAP and TSPEC is managed by hybrid coordination function (HCF). HCF has two modes of operation, enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA). EDCA is a contention-based channel access function and operates alternately with HCCA that is based on a centralized polling mechanism. The polling mechanism is controlled by the hybrid coordinator (HC) that is co-located with the quality of service access point (QAP). The HC performs bandwidth management including the allocation of transmission opportunity (TXOP) to QoS wireless nodes (QSTAs). A TXOP obtained using the contention-based channel access, it is called an EDCA-TXOP while a TXOP granted through HCCA is called a HCCA-TXOP or a polled TXOP [3]. The duration of the EDCA-TXOP is distributed to non-AP QSTAs in the beacon frames along with other EDCA related parameters.

The EDCA mechanism provides differentiated, distributed access to the wireless medium (WM) for QSTAs using eight different user priorities (UPs). The EDCA mechanism defines four access categories (ACs), AC_VO (for voice traffic), AC_VI (for video traffic), AC_BE (for best-effort traffic) and AC_BK (for background traffic). AC_VO possesses the highest priority and AC_BK has the lowest. Each AC has its own queue and parameter set. The EDCF parameter set includes *Minimum Contention Window Size* (CW_{min}), *Maximum Contention Window size* (CW_{max}), *Arbitration Inter-Frame Space (AIFS),* and *Transmission Opportunity limit (TXOPlimit).* CW is set as CW_{min} at the very beginning. A successful transmission will reset CW to CW_{min}. Instead of a DIFS, a wireless node needs to defer for AIFS. The ACs is derived from the user priorities (UPs). The differentiation in priority between the ACs is realized by setting different values for the AC parameters, which are arbitrary inter-frame space number (AIFNS), contention window size and transmission opportunity (TXOP) limit. Figure 1 shows the inter-frame space relationship used in DCF and EDCF.

HCCA uses polling access to the wireless medium and QoS polling can take place during both CFP and CP. The central concept of HCCA is controlled access phase (CAP), that is a bounded time interval and formed by a concatenating series of HCCA TXOPs. Scheduling of HCCA TXOP and formation of CAP are performed by the HC. When the HC needs access to the wireless medium (WM) to start a CFP or a TXOP in CP,

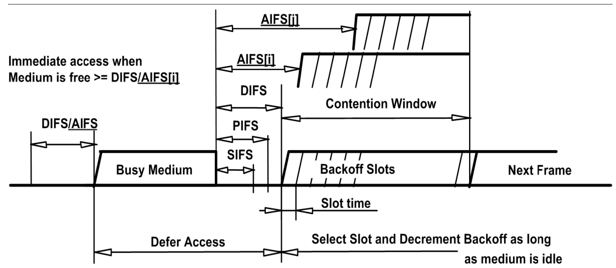

Fig. 1. Inter-frame space relationship in DCF and EDCF.

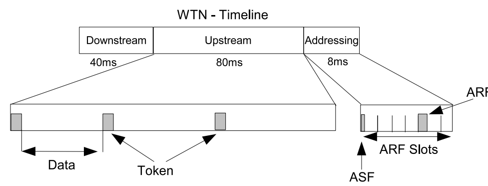

Fig. 2. The time division in WTN.

the HC senses the WM to determine if it is idle for a PIFS period. The HC then transmits the first frame of any permitted frame exchange sequence, with the duration value set to cover the CFP or the TXOP. The first permitted frame in a CFP after a target beacon transmission time (TBTT) is the beacon frame [3].

C. Wireless Token Network Protocol (WTN)

Wireless Token Network (WTN) protocol is a centralized polling MAC protocol. It is designed to improve performance of real-time traffic in WLAN by lowering transmission overhead and reducing packet loss due to collisions [6]. Specifically, we have reduced the size of the frames and subframes and provided time division multiplexing (TDM) to avoid collisions. All management functions are carried out in the Access Point (AP) and a wireless node can only send when they receive the token from the AP.

There are three time division multiplexed activities in each WTN cycle, addressing, downstream traffic and upstream traffic as shown in Figure 2. In a round robin cyclical network, the average delay experienced by each node is capped at approximately the cycle time. The delay requirements of conversation voice and video traffic have an upper bound of 150 ms and 200 ms respectively [7]. The data traffic is treated as the least priority traffic and therefore does not have an upper bound constrain. It uses any available bandwidth not used by either voice or video traffic.

The cycle is either set to 128 ms when addressing takes place or to 120 ms when addressing does not take place. This cycle time is divided between 40 ms downstream, 80 ms upstream and 8 ms addressing. Due to the lower overhead of downstream traffic, a much shorter time is allocated to it than upstream traffic. A cycle starts with downstream traffic from the access point. This traffic is sent continuously until either the AP runs out of traffic or its downstream period expires. This is significantly different from most schemes, where the

AP needs to compete like any other station to have access to the channel. Instead there is a more symmetrical traffic pattern without a bottleneck at the AP for received traffic.

After the downstream time division is completed, the upstream sequence commences, where tokens are passed to each client in turn. The token contains information regarding the duration the client can transmit for each slot allocated to it. This time can be fully utilized, or if a node runs out of traffic, it send a small empty packet to indicate that it is relinquishing the slot. After the upstream traffic time division is completed, the access point checks to see if a free address is available. If an address is available, addressing procedure takes place. Once the addressing procedure is completed, the cycle repeats. If no address is available then addressing procedure can be omitted and the cycle repeats.

WTN provides both the AP and all clients a dual queue system that allows best effort and real-time traffic to be separated and hence differentiated to provide QoS. By differentiating traffic at the node and giving the appropriate time slice to each client, stringent QoS can be obtained in terms of throughput for each stream. During the upstream sequence each client embeds information about the changes in its queue lengths in the data frames that are being sent. This information is stored in the management list.

To allow new wireless nodes in the network to associate with the network's AP, an addressing time is allocated. At the end of the upstream time division, the AP monitors for a free addressing and if one is found it sends an Address Send Frame (ASF). This signals to the unassociated nodes that an addressing period has begun and an address is available to any new wireless nodes intending to associate. To prevent collisions during associating with the AP and to provide fairness, all new nodes need to apply a random backoff slot before transmitting an Address Reply Frame (ARF). The first ARF received by the AP wins the contention and the address is given to the winning node. If no free address is available, the AP commences a downstream division without sending any ASF.

II. Simulation Scenario

Our main objectives for the simulation are to investigate the throughput performance of converging voice, video and data traffic in the legacy 802.11 and 802.11e and to compare them with our WTN protocol. We create a wireless network scenario with each wireless node transmitting either voice, video or data traffic to a single AP. We begin with three different wireless nodes sending voice, video and data traffic. We increase the number of nodes for each type of traffic in our simulation and monitor the throughput of each traffic type received by the AP. The simulation scenario is shown in Figure 3.

In line with the traffic characteristics used in real wireless network environments and digitized with the G.711 coding standard, the inter-arrival time of voice traffic is made 20 msec with a packet size of 160 bytes [7]. For video traffic the inter-arrival time is 10 msec with a packet size of 1280 bytes. The

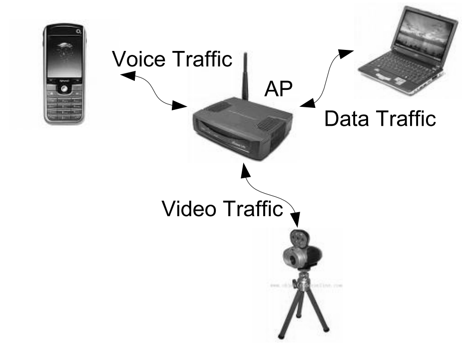

Fig. 3. Simulation scenario.

TABLE I
Parameters of DCF Used in Simulations.

Parameter	SIFS	DIFS	Slot Time	CW_{min}	CW_{max}
802.11b PHY	10 μsec	50 μsec	20 μsec	31	1023

data traffic, which is a best effort service is set with an inter-arrival time of 1.5 msec and a packet size of 500 bytes. With these parameters, the data rate for voice and video are 64 kbps and 1.024 Mbps respectively. User Datagram Protocol (UDP) is used for voice and video traffic while Transport Control Protocol (TCP) is used for best-effort traffic. Our simulations use the MAC protocol parameters from the IEEE standards [2] [3] as shown in Tables I and II.

The transmission rate is set at 11 Mbps for all the simulated protocols. The mean throughput of each traffic is measured in each simulation scenario which provides a snapshot of QoS performance for the converging traffic. We use ns2 [8] as our simulation tool and plot the mean throughput and access delay of each traffic received by the AP. We conduct several runs of simulations and achieved a confidence intervals of more than 95%.

III. Simulation Results

Our results show that our proposed WTN protocol outperforms the legacy 802.11b and 802.11e in a converging traffic for all voice, video and best-effort traffic. Figure 4 shows that the mean throughput of voice traffic of 802.11b protocol starts to drop voice packets at 12 nodes. As the number of nodes increases, more packets are being dropped. This is due to high packet collision rates and packets drops at the queue of each wireless node for the 802.11b protocol. For 802.11e protocol, voice traffic throughput is maintained until 18 nodes. This is equivalent to 6 voice nodes, 6 video nodes and 6 best-effort

TABLE II
Parameters of EDCF Used in Simulations.

Traffic	Transport Protocol	CW_{min}	CW_{max}	AIFNS
Voice	UDP	3	7	30 μsec
Video	UDP	7	15	30 μsec
Data	TCP	15	1023	50 μsec

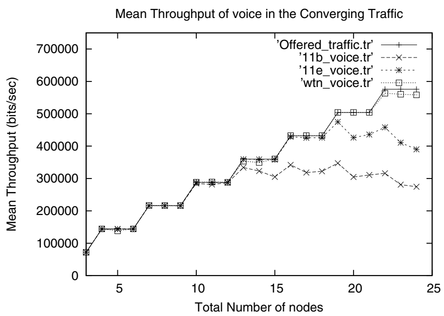

Fig. 4. Mean Throughput of Voice Traffic.

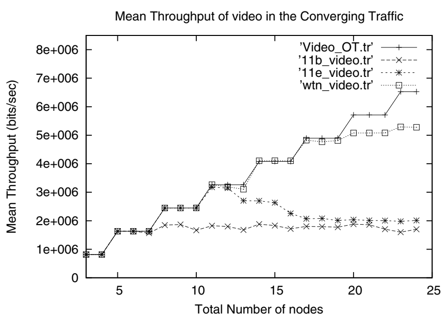

Fig. 5. Mean Throughput of Video Traffic.

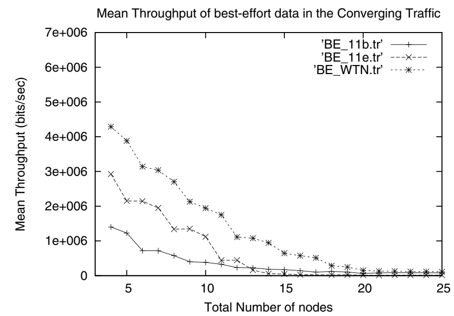

Fig. 6. Mean Throughput of Best-Effort Traffic

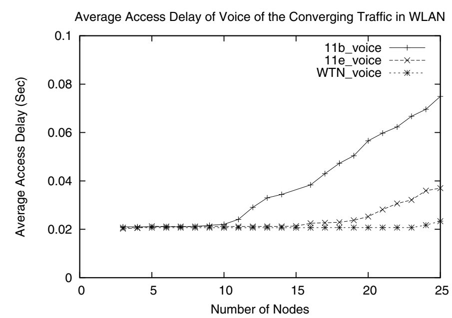

Fig. 7. Average Access Delay of Voice Traffic in the Converging Traffic in WLAN.

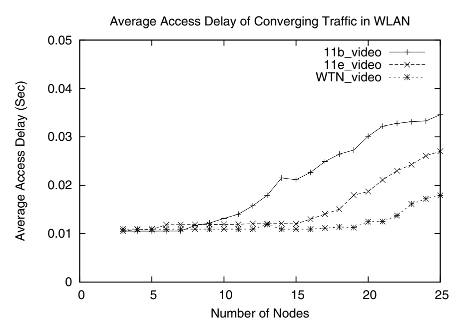

Fig. 8. Average Access Delay of Video Traffic in the Converging Traffic in WLAN.

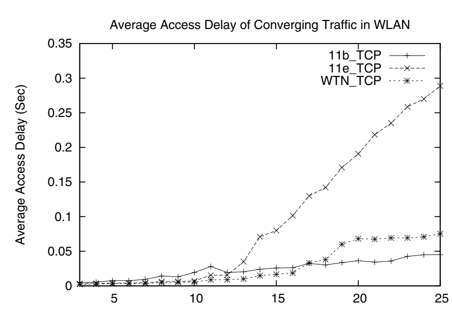

Fig. 9. Average Access Delay of Best-Effort Data in the Converging Traffic in WLAN.

nodes in a single WLAN basic service set (BSS). The WTN MAC protocol performs better than both the legacy 802.11b and 802.11e. It manages to maintain voice traffic throughput up to 22 nodes of converging traffic without being dropped which is equivalent to 7 of each voice, video and best-effort traffic in a single WLAN BSS.

The performance of video traffic in a converging data is shown in Figure 5. It is shown that the legacy 802.11b supports 8 nodes in a single WLAN BSS without dropping video packets. Beyond that number, nodes start to lose video packets. In comparison, 802.11e supports up to 13 nodes before video packets start dropping. With WTN, 19 converging nodes are supported before video packets start to drop. The performance of the best-effort traffic using the transmission control protocol (TCP) in the converging traffic environment is shown in Figure 6. It is shown that the WTN protocol provides a significantly better throughput of best-effort traffic compared to 802.11e and the legacy 802.11b. With 4 nodes in the single BSS, best-effort traffic total throughput in WTN is about 4.3 Mbps while for 802.11e and 802.11b it is about 3 Mbps and 1.4 Mbps respectively. As the number of nodes is increased, the best-effort traffic of WTN maintains higher throughput than 802.11e and 802.11b.

Figures 7, 8 and 9 show the average access delay of voice, video and best-effort traffic respectively. Average access delay of voice traffic in WTN is consistence at about 0.02 sec and starts to increase at 25 nodes. In the 802.11e, voice traffic maintains the average access delay at below 0.02 sec until the number of nodes reach 22, where average access delay increases linearly. Average access delay of voice traffic in

802.11b is about 0.02 sec for number of nodes less than 10. Beyond 10, the delay increases steadily and at 25 nodes, delay of more than 70 msec is observed. Video traffic delay for 10 nodes in the WTN is about 11 msec while for 802.11e and 802.11b is about 12 msec and 14 msec respectively. For best-effort traffic using TCP protocol, the highest delay is observed in the 802.11e for nodes more than 12, while the lowest delay for best-effort traffic is in the 802.11b. While in WTN, at 25 nodes access delay is about 90 msec.

In the environment where three main traffic, voice, video and data converge in WLAN, WTN supports 4 voice traffic nodes more than 802.11e and 10 more than the legacy 802.11b with sustainable throughput for all traffic. For video traffic, WTN supports 6 video traffic nodes more than 802.11e and 11 more than the legacy 802.11b. Significantly higher throughput of best-effort traffic in WTN is shown compared to 802.11e and the legacy 802.11b. Clearly, a degree of fairness to all types of traffic is achieved in WTN while implementing prioritization to real-time traffic. As shown in Table III, 24 nodes of voice traffic is supported in the WTN compare to 18 nodes and 12 nodes for the 802.11e and the legacy 802.1b respectively. As for video traffic, 19 nodes are supported by the WTN, 13 nodes by the 802.11e and 8 nodes by the 802.11b.

The better performance shown with 802.11e compared to legacy 802.11b with real-time traffic is because of its prioritization and differentiation mechanism implemented in its protocol for real-time traffic which supports findings by other researchers [9] [10] [11]. The average access delay in WTN protocol is more consistence than the 802.11e and the legacy 802.11b. As WTN is based on centralized polling, each nodes will access the medium at the deterministic interval. This will provide easier management of different traffic types in WLAN and also easier implementation of admission control. Collision of packets is minimized in WTN as each node can only transmit when a token is received. This provide a maximum utilization of bandwidth and reduce losses of packets.

IV. Conclusion

This work has investigated the performance of converging traffic in WLAN for the legacy 802.11 and 802.11e and has proposed a technique to optimize MAC protocols by reducing overhead and collision of packets in WTN protocol. It has been shown by using the WTN protocol, significant improvement in terms of throughput and delay has been achieved. More real-time traffic with QoS can be supported in a single WLAN BSS as compared to the legacy 802.11 and 802.11e protocols. The limited resources of WLAN are optimized and throughput performance is enhanced in this protocol.

In our simulation results, we showed that prioritization and differentiation mechanism used in the 802.11e has its drawback, in that it suppresses best-effort traffic as seen in Figure 6. With more than 14 nodes in a single BSS, best-effort traffic has nearly zero throughput in 802.11e while 802.11b and WTN sustain significantly higher throughput. In WTN, all three types of traffic are served in a fairly manner even though prioritization of voice and video traffic is implemented.

TABLE III

Comparison on the Number of Nodes Supported with QoS in a Single WLAN BSS of the Three Simulated Protocols

	Voice traffic	Video Traffic	Best-effort traffic
WTN	24 nodes	19 nodes	15 nodes
802.11e	18 nodes	13 nodes	10 nodes
802.11b	12 nodes	8 nodes	5 nodes

While providing higher throughput for real-time traffic, best-effort traffic using TCP protocol is not suppressed in a manner 802.11e does. The best-effort traffic is sustained up to 15 nodes with total mean throughput of 900 Mbps.

Although our works have shown that significant improvements on throughput and delay have been achieved on the convergence traffic in WLAN, further work is needed to provide guaranteed QoS for real-time traffic. Extending the work on WTN, our future works will include implementation of more efficient timing schedule at each node, admission control mechanism and bandwidth reservation.

References

[1] Colubris Networks, “The Intelligent Wireless Networking Choice - Converging Voice, Video and Data Onto a Single, Scalable WLAN Network”, 2005. http://www.colubris.com.

[2] IEEE, “Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications,” 1999.

[3] IEEE, “Part 11: Wireless LAN Medium Access Control (MAC) and Physical layer (PHY) specifications, Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements”, 11 November 2005.

[4] Bianchi, “Understanding 802.11e contention-based prioritization mechanism and their coexistence with legacy 802.11 stations”, IEEE Network, pp.28-34, 2005.

[5] Deng and Yen, “QoS provisioning system for multimedia transmission in IEEE 802.11 WLAN”, IEEE J. Sel. Areas Comm. 23, No.6, pp.1240-1252, 2005.

[6] Justin Wyatt, Daryoush Habibi, Iftekhar Ahmad and Hushairi Zen, “Providing QoS for Symmetrical Voice/Video Traffic in Wireless Networks” accepted by 15th IEEE International Conference on Networks, ICON 2007.

[7] I.I.E Consortium, “WLAN network capacity analysis,” 2006.

[8] K.Fall and K. Varadhan, “The ns manual,” 2005.

[9] Nitin Gupta and P.R. Kumar, “A Performance Analysis of The 802.11 Wireless LAN Medium Access Control”, Communications in Information and Systems, Vol.3, No. 4, pp. 279-304, September 2004.

[10] P.Chatzimisios, A.C. Boucouvalas and V.Vitsas, “Performance analysis of the IEEE 802.11 MAC protocol for Wireless LANs”, International Journal of Communication Systems, No.18, pp 545-569, 2005.

[11] Peter P.Pham, “Comprehensive Analysis of the IEEE 802.11”, Mobile Networks and Applications, No 10, pp 691-703, 2005.

[12] Aura Ganz, Zvi Ganz, and Kitti Wongthavarawat, "Multimedia Wireless Networks - Technologies, Standards, and QoS", Prentice Hall, Upper Saddle River, 2004.